Translation of Japanese Unexamined Patent Application

METHOD OF GUARANTEEING VOICE QUALITY IN AN INTERNET TELEPHONY SERVICE

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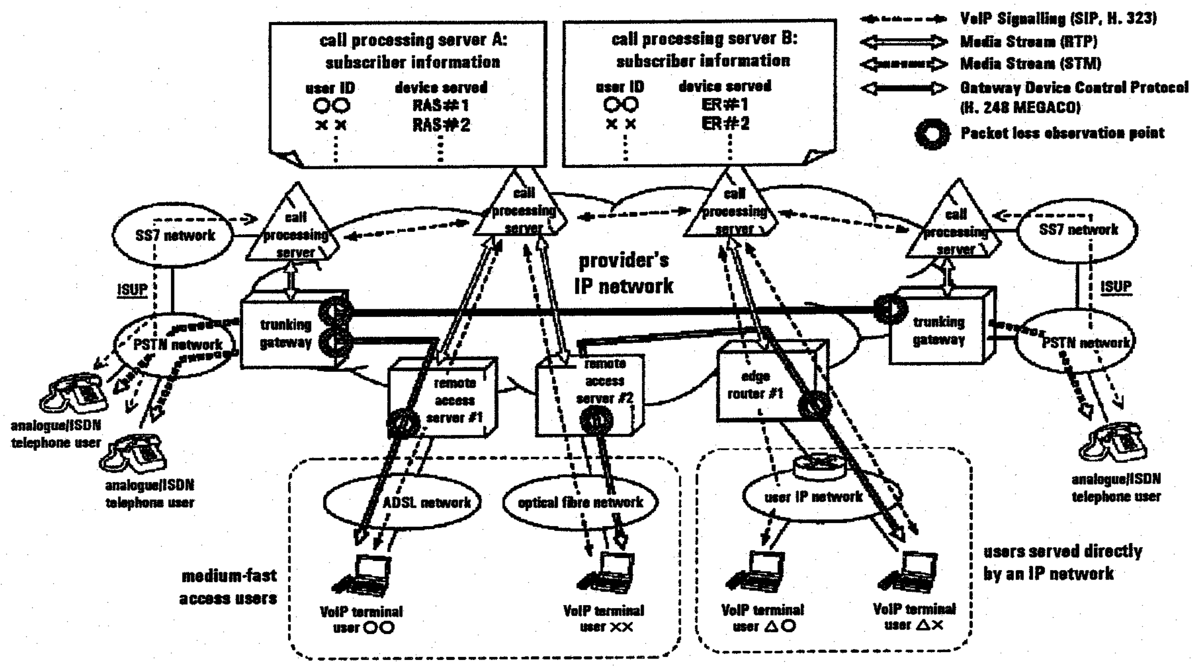
Abstract



TASK: To enable an IP network provider offering an Internet telephony service to provide billing based on guaranteed call quality.

SOLUTION: Packet loss per unit time of each RTP session is observed at a gateway device disposed at the boundary between a provider's IP network and a user access network, and if a threshold value (which depends on the codec used) held by the gateway device or given by a call processing server is exceeded, the call processing server is notified, whereupon the call processing server reacts by either making the call (or the user) using the RTP session in question non-billable or forcing disconnection, so that a time-based charge is not generated when voice quality has deteriorated. The threshold value may be determined empirically by making an objective measurement of voice quality (as typified by R-value, PSQM, etc.) and correlating the state of deterioration of voice quality due to packet loss rate with the codec and packetizing period.

Claims

1. A method of guaranteeing voice quality in an Internet telephony service, characterised in that in the case of a provider such as an ISP or an ASP who arranges, on the provider's IP network that uses an Internet protocol, call processing servers such as for example SIP proxy servers or gatekeeper servers having a function for converting between the user ID and the IP address of a call processing message in an Internet telephony protocol such as for example SIP or ITU-T H.323, and who offers an Internet telephony service to users who utilize a protocol such as IP or IP over PPP to connect to the provider's IP network:

when such a provider implements billing by calling time, the provider guarantees a certain minimum voice quality for each call, and if call quality falls below standard, the provider stops billing or forces disconnection, from the network side, of the call in question.

2. The method of guaranteeing voice quality in an Internet telephony service as set forth in Claim 1, wherein gateway devices arranged at connection points in all access modes between said provider's IP network and either a user IP network or an IP over PPP terminal, monitor – under the control of a call processing server – packet flows from the provider's IP network to users during RTP sessions specified by source and destination IP addresses and source and destination port numbers, and notify a call processing server if a packet loss occurring per unit time in the provider's IP network reaches or exceeds a threshold value.

3. The method of guaranteeing voice quality in an Internet telephony service as set forth in Claim 1 or 2, wherein a call processing server that has received a said notification immediately performs a procedure to stop the billing of the RTP session in question, or forces disconnection of the call that is using the RTP session in question.

4. The method of guaranteeing voice quality in an Internet telephony service as set forth in any one of Claims 1 to 3, wherein said threshold value is either held by a said gateway device or is set for each RTP session by an instruction from a call processing server; and, by taking into account the codec – and, if possible, the packetizing period – which the RTP session in question is using, different values can be used for said threshold according to circumstances.

5. The method of guaranteeing voice quality in an Internet telephony service as set forth in any one of Claims 1 to 4, wherein said threshold value is determined in accordance with codec and packetizing period, on the basis of the correlation between packet loss rate and the result of objective [1]* measurement of voice quality based on R-value, PSQM, etc.

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* Numbers in square brackets refer to Translator's Notes appended to the translation.

6. The method of guaranteeing voice quality in an Internet telephony service as set forth in any one of Claims 1 to 5, wherein the point at which said call processing server is notified if said packet loss rate reaches or exceeds the threshold value is when, for at least a prescribed period of time, no packet of the RTP session in question has been observed after a prescribed period of time has passed; and under these circumstances a call processing server forces disconnection of the call that is using the RTP session in question.

7. The method of guaranteeing voice quality in an Internet telephony service as set forth in any one of Claims 1 to 6, wherein when said gateway device actually terminates an RTP session upon observing said packet loss, it does not only measure the packet loss rate at the IP level, but also takes delay jitter into consideration and uses, as an index, the packet loss rate when [packets] are actually recovered as voice. [2]

Detailed Description of the Invention

Technical field of the invention

[0001] The present invention relates to Internet telephony, and in particular to methods of guaranteeing voice quality when time-based billing is carried out in an Internet telephony service.

Prior art

[0002] Internet telephony enables communication between personal computers in the form of IP packets, provided that the PCs in question are connected to microphones and speakers, equipped with voice communication software, and connected to the Internet. Internet telephony also enables voice communications between a personal computer and a conventional telephone by assigning, to a gateway device, the processing by the personal computer. An IP network provider provides services for this internet telephony.

Problems which the invention is intended to solve

[0003] When an IP network provider offers an Internet telephony service using call processing servers owned by the provider in a multi-QoS network likewise owned by the provider, it has hitherto been difficult – for reasons (1) to (3) listed below – to guarantee call quality for Internet telephony users.

[0004] (1) Given the nature of the Internet Protocol (IP), the Real-Time Transport Protocol (RTP) used for voice communications and the User Datagram Protocol (UDP), there is no concept of end-to-end connection or of node-to-node connection in IP networks, and it has not been possible to guarantee normal data communications.

[0005] (2) For the reason given in item (1) above, if – due to factors internal to a network – there has been very significant loss of voice packets from an Internet telephony user who is making a call, it is not been possible for a service provider to grasp the situation in real time. As a result, when billing on the basis of calling time, billing continues unabated even in circumstances where a user cannot carry out voice communication due to some provider-side factor.

[0006] (3) Although conventional IP routers are provided with a function for collecting statistical information (packet loss, etc.) at the session level, it has not been possible to support this in a Voice over IP (VoIP) service, where the session information to be observed (sent and received IP addresses, sent and received port numbers) changes dynamically with each call.

[0007] (4) The packet loss rate which can be collected by a conventional IP router is collected simply as an index of network quality, and cannot be used just as it is as an index of voice quality in an Internet telephony service. Moreover, packet loss rate has had to be controlled by means of codec and packetizing period, taking into account the effect of voice quality on packet loss. [3]

[0008] It is an object of the present invention to provide a method of guaranteeing voice quality whereby an IP network provider offering an Internet telephony service is able to provide billing based on guaranteed call quality.

Means for solving problems

[0009] To solve the problems described above, the present invention has the following features. (1) Packet loss per unit time of each RTP session is observed at a gateway device (a router, a remote access server, a trunking gateway, etc.) disposed at the boundary between the provider's IP network and a user access network, and if a threshold value (which depends on the codec used) held by the gateway device or given by a call processing server is exceeded, the call processing server is notified.

[0010] (2) In the operation described in item (1) above, packet loss rate is measured only for packet flows from a gateway to a user.

[0011] (3) In the operation described in item (1) above, a call processing server that has been notified that a threshold value has been exceeded reacts by either making the call (or the user) using the RTP session in question non-billable or forcing disconnection, so that a time-based charge is not generated when voice quality has deteriorated.

[0012] (4) The threshold value mentioned in item (1) above is determined empirically by making an objective measurement of voice quality (typified by R-value, PSQM, etc.) and

correlating the state of deterioration of voice quality due to packet loss with the codec and packetizing period.

[0013] (5) A standard protocol is used to transfer, via the IP network, RTP session information (sent and received IP addresses, sent and received port numbers, codec used, threshold values, etc.) to gateway devices, and to notify call processing servers when a threshold has been exceeded.

[0014] (6) In the operation described in item (4) above [4], the Internet telephony user information which a call processing server holds as its own subscriber information is information indicative of which Internet telephony terminal is served by which gateway device. On this basis, a call processing server selects the relevant gateway device for each call.

[0015] (7) The point at which notification is given in item (1) above can be when, for at least a prescribed period of time, no packet of the RTP session in question has been observed after a prescribed period of time has passed.

[0016] (8) Call processing servers are provided with a function whereby, when the notification described in item (6) above [5] is received, the call processing server forces disconnection of the call that is using the RTP session in question.

[0017] (9) When a gateway device actually terminates an RTP session in the manner of a trunking gateway device (a device for implementing voice calls between an IP network and a telephone network (PSTN)) upon observing packet loss in item (1) above, instead of merely measuring packet loss rate at the IP level, it can also take delay jitter into consideration and use, as an index, the packet loss rate when [packets] are actually recovered as voice. [6] Accordingly, the method of the present invention has the following distinguishing features.

[0018] (Claim 1) In the case of a provider such as an ISP or an ASP who arranges, on the provider's IP network that uses an Internet protocol, call processing servers such as for example SIP proxy servers or gatekeeper servers having a function for converting between the user ID and the IP address of a call processing message in an Internet telephony protocol such as for example SIP or ITU-T H.323, and who offers an Internet telephony service to users who utilize a protocol such as IP or IP over PPP to connect to the provider's IP network: when such a provider implements billing by calling time, the provider guarantees a certain minimum voice quality for each call, and if call quality falls below standard, the provider stops billing or forces disconnection, from the network side, of the call in question.

[0019] (Claim 2) Gateway devices arranged at connection points in all access modes between the provider's IP network and either a user IP network or an IP over PPP terminal, monitor –

under the control of a call processing server – packet flows from the provider's IP network to users during RTP sessions specified by source and destination IP addresses and source and destination port numbers, and notify a call processing server if a packet loss occurring per unit time in the provider's IP network reaches or exceeds a threshold value.

[0020] (Claim 3) A call processing server that has received a notification immediately performs a procedure to stop the billing of the RTP session in question, or forces disconnection of the call that is using the RTP session in question.

[0021] (Claim 4) The threshold value is either held by a gateway device or is set for each RTP session by an instruction from a call processing server; and, by taking into account the codec – and, if possible, the packetizing period – which the RTP session in question is using, different values can be used for the threshold according to circumstances.

[0022] (Claim 5) The threshold value is determined in accordance with codec and packetizing period, on the basis of the correlation between packet loss rate and the result of objective measurement of voice quality based on R-value , PSQM, etc.

[0023] (Claim 6) The point at which the call processing server is notified if the packet loss rate reaches or exceeds the threshold value is when, for at least a prescribed period of time, no packet of the RTP session in question has been observed after a prescribed period of time has passed; and under these circumstances a call processing server forces disconnection of the call that is using the RTP session in question.

[0024] (Claim 7) When the gateway device actually terminates an RTP session upon observing the packet loss, it does not only measure the packet loss rate at the IP level, but also takes delay jitter into consideration and uses, as an index, the packet loss rate when [packets] are actually recovered as voice. [7]

Mode of practising the invention

[0025] A schematic network configuration is shown in FIG. 1. A service provider such as an Internet service provider (ISP) or an application service provider (ASP) arranges call processing servers such as SIP proxy servers (SIP) or gatekeeper servers (H.323) in the provider's IP network. These call processing servers use the Internet protocol (IP) and have a function for converting between the user ID and the IP address of a call processing message in an Internet telephony protocol as typified by SIP (Session Initiation Protocol: IETF RFC 2543) or ITU-T H.323. The service provider thereby provides an Internet telephony service to users who utilize a protocol such as IP or IP over PPP to connect to the provider's IP network.

[0026] The function of each part in the configuration depicted in FIG. 1 is as follows.

[0027] (1) The modes of access from a personal computer user or a conventional telephone user to the provider's IP network are assumed to be connection from an existing telephone network, connection by means of a medium-high speed access line such as ADSL or optical fibre, or direct accommodation of a user IP network via an Ethernet (registered trademark), a dedicated line, etc.

[0028] (2) Users are connected to the provider's IP network via various kinds of gateway device.

[0029] (3) A voice call between the IP network and a telephone network (PSTN) has the following form. Namely, in the case of Internet telephones and telephones, the call processing protocol (H.323, SIP, ISUP, etc.) is terminated by call processing servers [8], and the trunking gateway devices are controlled from the call processing servers by a standard protocol (here, ITU-T H.248/MEGACO is envisaged). Note that an alternative configuration in which the trunking gateway devices are integrated with the call processing servers is also feasible.

[0030] (4) The call processing servers hold, as subscriber information, at least (i) user IDs as stipulated by the Internet telephony protocol, and (ii) information relating to the gateway devices (IP address, etc.) serving those users. Note that in the case of connection to an existing telephone network, call processing servers hold information relating to the trunking gateway devices which serve the called telephone numbers (over STM channels).

[0031] (5) Call processing servers use a standard protocol (here, ITU-T H.248/MEGACO is envisaged) to control the gateway devices.

[0032] (6) Each type of gateway device is provided with a function for measuring packet loss per unit time of packet flows coming out into an access network, for RTP sessions stipulated by control from a call processing server.

[0033] (7) The call processing servers are provided with a time-based billing function for billing for services.

[0034] Given the configuration described above, the following processing is performed when SIP is used as the Internet telephony protocol and MEGACO is used as the protocol for gateway device control from the call processing servers. FIG. 2 and FIG. 3 show the call connection sequences involved.

[0035] FIG. 2 shows the sequence up to connection between users and start of the call. A destination user is called (step S1). The call processing servers look up the gateways serving the destination and source users (steps S2 and S3). The gateway devices (routers, remote

access servers, trunking gateways, etc.) disposed at the boundary between the provider's IP network and the user access networks start to observe packet loss rate of the RTP session in question and, depending on the codec used, hold within the gateway device [9] (steps S4 and S5), and the call starts when the destination goes off-hook (step S6).

[0036] FIG. 3 shows the sequence when congestion or a fault occurs in the provider's IP network, in which cases the following processing is performed.

[0037] At step S11, if the packet loss rate being observed by a gateway device exceeds the threshold given by a call processing server, the gateway device notifies the call processing server of the deterioration in voice quality by mapping it to a Quality Alert message.

[0038] Note that it is sufficient to measure packet loss rate in respect of just the packet flow from the gateway to the user. Moreover, when a gateway device actually terminates an RTP session upon observing packet loss, in the manner of a trunking gateway device (a device for implementing voice calls between an IP network and a telephone network (PSTN)), instead of merely measuring packet loss rate at the IP level, it can also take delay jitter into consideration and use, as an index, the packet loss rate when [packets] are actually recovered as voice. [10]

[0039] The threshold value is determined empirically by making an objective measurement of voice quality (as typified by R-value, PSQM, etc.) and correlating the state of deterioration of voice quality due to packet loss rate with the codec and packetizing period.

[0040] A standard protocol is used to transfer, via the IP network, RTP session information (sent and received IP addresses, sent and received port numbers, codec used, threshold values, etc.) to gateway devices, and to notify call processing servers when a threshold has been exceeded.

[0041] The point at which notification is given can also be when, for at least a prescribed period of time, no packet of the RTP session in question has been observed after a prescribed period of time has passed.

[0042] At step S12, a call processing server that has been notified that a threshold value has been exceeded reacts by either making the call (or the user) using the RTP session in question non-billable or forcing disconnection, so that a time-based charge is not generated when voice quality has deteriorated.

[0043] Note that the Internet telephony user information which a call processing server holds as its own subscriber information is information indicative of which Internet telephony terminal is served by which gateway device. On this basis, the call processing server selects

the relevant gateway device for each call. Call processing servers can also be provided with a function which, when a notification has been received, forces disconnection of the call that is using the RTP session in question.

Benefits of the invention

[0044] As has been described above, the present invention provides an Internet telephony service wherein, when implementing billing by calling time and guaranteeing a certain minimum voice quality for each call, if call quality falls below standard, it is possible to stop billing or to force disconnection, from the network side, of the call in question, so that a charge does not continue to be generated when voice quality has deteriorated.

Brief Description of the Drawings

FIG. 1 schematises the network structure of an Internet telephony service, and depicts a mode of practising the present invention.

FIG. 2 shows the call connection sequence up to connection between users and the start of a call in this mode of practising the invention.

FIG. 3 shows the sequence when congestion or a fault has occurred in the provider's IP network in this mode of practising the invention.

FIG. 1

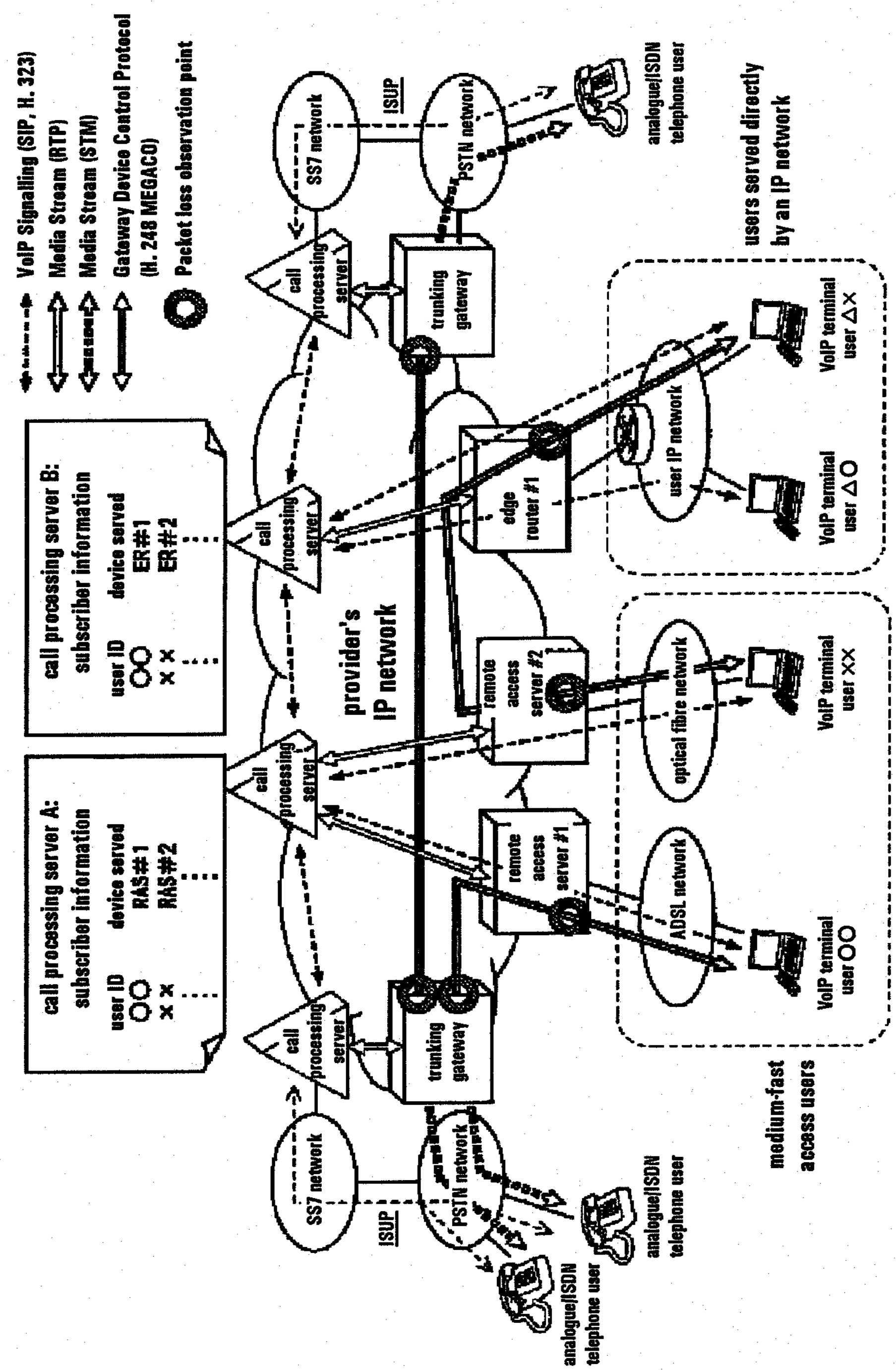

VoIP Signalling (SIP, H. 323)
Media Stream (RTP)
Media Stream (STM)
Gateway Device Control Protocol (H. 248 MEGACO)
Packet loss observation point
call processing server A: subscriber information
user ID device served
OO RAS#1
× × RAS#2
call processing server B: subscriber information
user ID device served
OO ER#1
× × ER#2
call processing server
provider's IP network
trunking gateway
remote access server #1
remote access server #2
edge router #1
SS7 network
PSTN network
ISUP
analogue/ISDN telephone user
ADSL network
optical fibre network
user IP network
VoIP terminal user OO
VoIP terminal user ××
VoIP terminal user △O
VoIP terminal user △×
medium-fast access users
users served directly by an IP network

FIG. 2

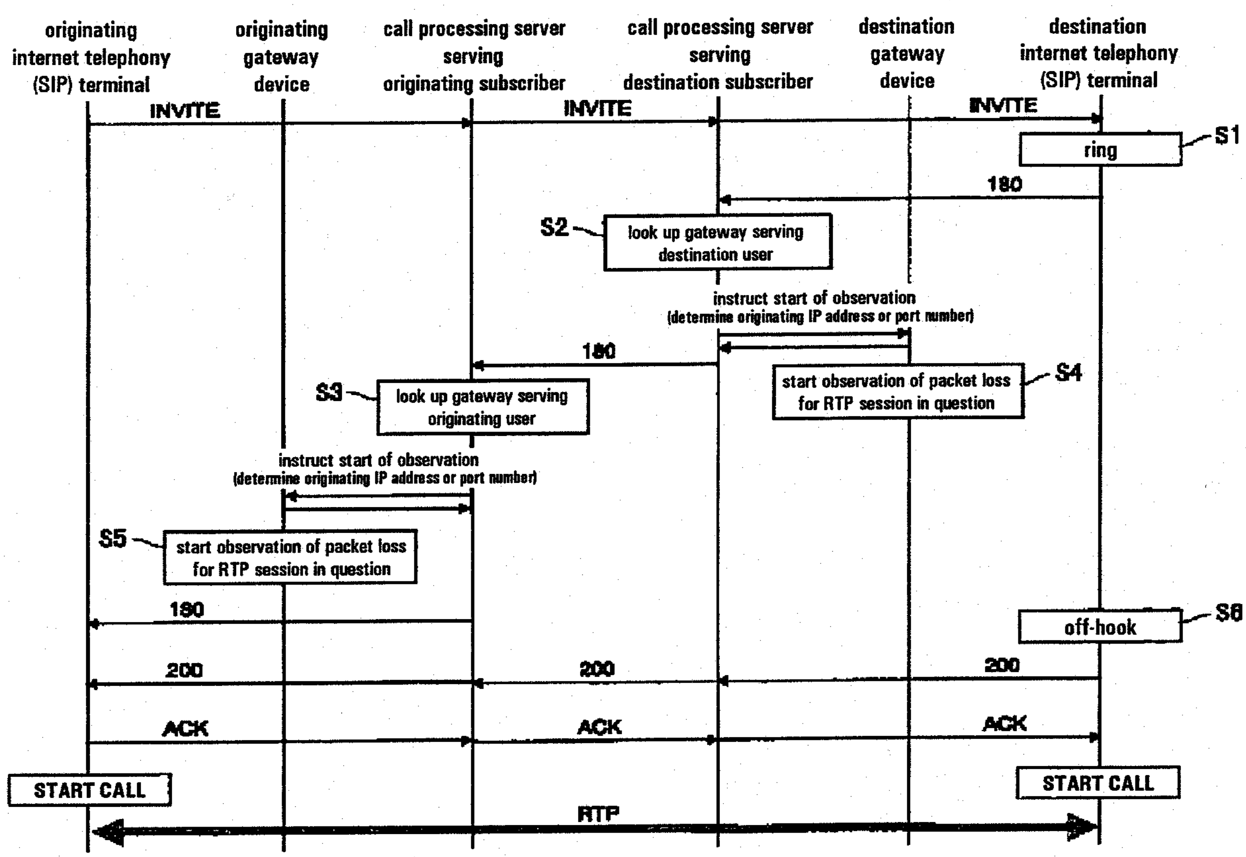
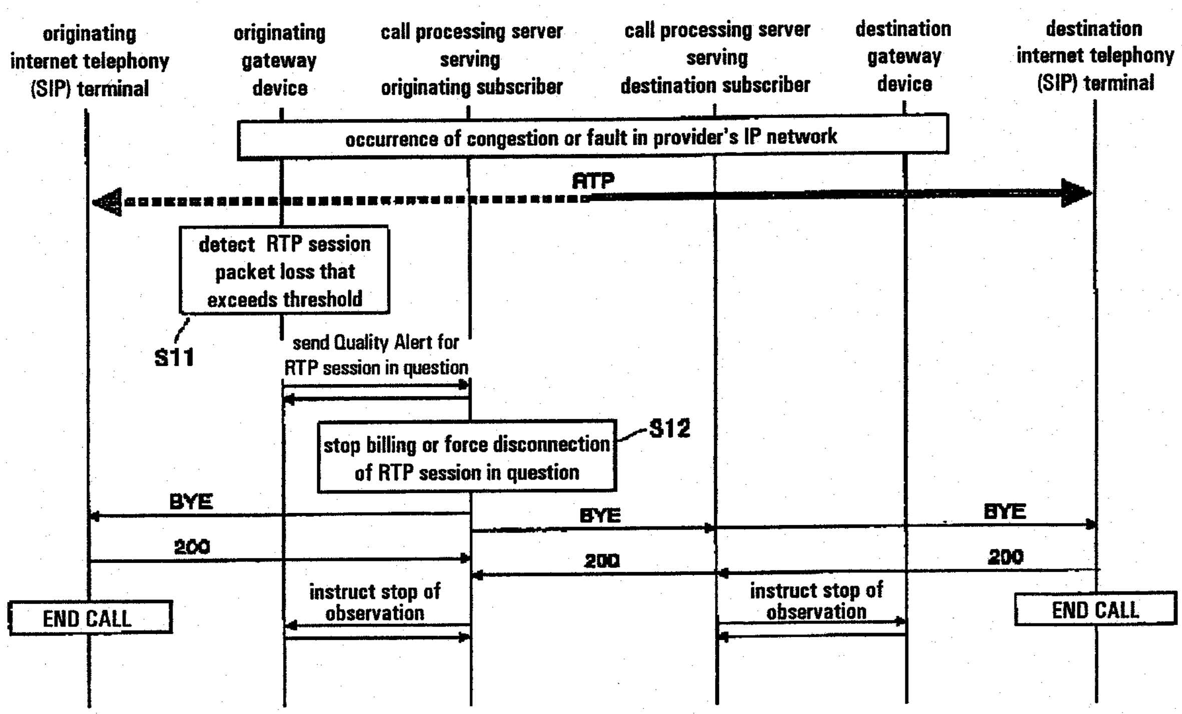

originating internet telephony (SIP) terminal
originating gateway device
call processing server serving originating subscriber
call processing server serving destination subscriber
destination gateway device
destination internet telephony (SIP) terminal
INVITE
INVITE
INVITE
ring
S1
180
S2
look up gateway serving destination user
instruct start of observation (determine originating IP address or port number)
180
start observation of packet loss for RTP session in question
S4
S3
look up gateway serving originating user
instruct start of observation (determine originating IP address or port number)
S5
start observation of packet loss for RTP session in question
180
off-hook
S6
200
200
200
ACK
ACK
ACK
START CALL
START CALL
RTP

FIG. 3


originating internet telephony (SIP) terminal
originating gateway device
call processing server serving originating subscriber
call processing server serving destination subscriber
destination gateway device
destination internet telephony (SIP) terminal
occurrence of congestion or fault in provider's IP network
RTP
detect RTP session packet loss that exceeds threshold
S11
send Quality Alert for RTP session in question
stop billing or force disconnection of RTP session in question
S12
BYE
BYE
BYE
200
200
200
END CALL
instruct stop of observation
instruct stop of observation
END CALL

TRANSLATOR'S NOTES

1. Sic.
2. I have added the word in square brackets to make sense of the Japanese, which omits any corresponding word.
3. Sic. Surely this is erroneous for "taking into account the effect of packet loss on voice quality".
4. Sic. I wonder if "item (4)" is erroneous and should have been "item (5)".
5. Sic. Surely this should be "the notification described in item (7) above".
6. See Note 2 above.
7. See Note 2 above.
8. Sic.
9. Sic. The Japanese sentence becomes strange at this point. It would have made technical sense as follows: "The gateway devices ... start to observe packet loss rate of the RTP session in question (steps S4 and S5)". The extra, problematic words, "and, depending on the codec used, hold within the gateway device" seem to be a reference to the codec-dependent thresholds held in the gateway devices. One way of construing the odd phrase is to conclude that crucial words have been erroneously omitted from the Japanese text at this point.
10. See Note 2 above.